AD-A112 008 NAVAL RESEARCH LAB . WASHINGTON DC F/G 20/1

A PRONY MEASURING SYSTEM FOR UNDERWATER ACOUSTICAL MEASUREMENTS--ETC(U)

JAN 82 C K BROWN, R W LUCKEY

UNCLASSIFIED NRL-MR-4735 NL

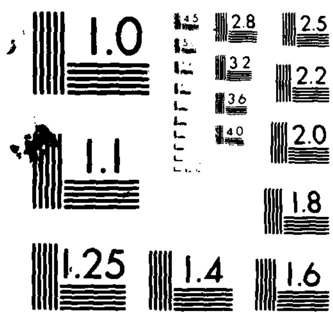
1 of 1

END

DATE FILMED

4 82

DTIC



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS 1963-A

UNCLASSIFIED

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

REPORT DOCUMENTATION PAGE		READ INSTRUCTIONS BEFORE COMPLETING FORM
1. REPORT NUMBER NRL Memorandum Report 4735	2. GOVT ACCESSION NO. AD-A112 008	3. RECIPIENT'S CATALOG NUMBER
4. TITLE (and Subtitle) A PRONY MEASURING SYSTEM FOR UNDERWATER ACOUSTICAL MEASUREMENTS		5. TYPE OF REPORT & PERIOD COVERED Interim report on a continuing problem. 6. PERFORMING ORG. REPORT NUMBER
7. AUTHOR(s) C.K. Brown and R.W. Luckey		8. CONTRACT OR GRANT NUMBER(s)
9. PERFORMING ORGANIZATION NAME AND ADDRESS Underwater Sound Reference Detachment Naval Research Laboratory PO Box 8337, Orlando, FL 32856		10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS PE: 63562N Task: S0221-AS WU: 0595-0
11. CONTROLLING OFFICE NAME AND ADDRESS Naval Ship R&D Center (Code 1908) Carderock Laboratory Bethesda, MD 20084		12. REPORT DATE January 31, 1982 13. NUMBER OF PAGES
14. MONITORING AGENCY NAME & ADDRESS(if different from Controlling Office)		15. SECURITY CLASS. (of this report) 15a. DECLASSIFICATION/DOWNGRADING SCHEDULE

16. DISTRIBUTION STATEMENT (of this Report)

Approved for public release; distribution unlimited

17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)

18. SUPPLEMENTARY NOTES

19. KEY WORDS (Continue on reverse side if necessary and identify by block number)

Acoustical measurements
Prony method
Analog-to-digital converter
Transient recorder
Computer control
Digital signal processing
General Purpose Interface Bus

20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

A new digital system, the Prony Measuring System, for underwater acoustical measurements from 1 Hz to approximately 2 MHz has been developed at the Underwater Sound Reference Detachment of the Naval Research Laboratory. The heart of the system is a stable signal and sampling source and 12-bit, 5-MHz, analog-to-digital converter with a 4K buffer memory. A PDP-11/23 serves as a controller as well as a signal processor for averaging, and amplitude and phase estimation. The measuring system as a whole can be

(over)

DD FORM 1 JAN 73 1473 EDITION OF 1 NOV 65 IS OBSOLETE
S/N 0102-LF-014-6601

UNCLASSIFIED

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

viewed as an instrument and can be controlled by a host computer or desk-top calculator via the IEEE-488 General-Purpose Interface Bus (GPIB).

TABLE OF CONTENTS

A PRONY MEASURING SYSTEM FOR UNDERWATER ACOUSTICAL MEASUREMENTS

INTRODUCTION

A new digital measuring system employing a buffered video analog-to-digital converter (ADC) has been developed at the Underwater Sound Reference Detachment of the Naval Research Laboratory (NRL-USRD). Called the PMS-01, the system name is derived from the words Prony Measuring System Number 1. This system was designed primarily to perform acoustical measurements using the Prony Method [1] [2]. This method derives frequency response information from transient characteristics of transducer signals. However, the PMS-01 can perform conventional measurements using swept-frequency signals equally well. The system is housed in a single rack and can be controlled by a computer or desk-top calculator via the popular IEEE-488 General-Purpose Interface Bus (GPIB). The system is pictured in Fig. 1.

The PMS-01 is a complete acoustical measuring system for fast, accurate calibration of underwater instruments. It contains a programmable signal source which provides a pulsed or continuous drive to a sound projector; a programmable multiplexer and receive amplifier to condition the resulting hydrophone output; a 12-bit video ADC and 4096-word memory to digitize and store the hydrophone output signal; and a controller which accepts this digitized data and communicates with external devices via the GPIB. A block diagram of the system is shown in Fig. 2.

HISTORY

The evolution of measuring systems at USRD has paralleled the evolution of electronic components and instrumentation in general. Early systems tended to be primarily analog and required a significant amount of attention from the operator. Later digital technology led to greater automation and precision; systems required fewer adjustments and produced more repeatable results. The introduction of computers allowed measurement sequences to be programmed with automatic reduction of the acquired data. This was enhanced by the development of efficient digital signal processing software. Computers became smaller and less expensive. Integrated circuits became more dense and operated at ever faster speeds.

All of these technological advances have led to the system configuration represented by the PMS-01: a programmable signal source, transient recorder, and microcomputer. These three components make up a truly universal measurement system for underwater acoustics.

SYSTEM SPECIFICATIONS

Signal Source:

Sine output frequency	1 Hz to 3 MHz
Pulse output frequency	1 Hz to 10 MHz
Frequency resolution	1 Hz
Operating modes	cw or pulse
Burst widths	1 μs to 10 s
Time resolution (nominal)	1 μs

Signal Receiver:

Bandwidth	1 Hz to 10 MHz
Gain	0 to 60 dB
Gain resolution	1-dB steps
Number of input channels	8
Input impedance	CH 0-4: 10 MΩ CH 5-7: 1 MΩ

Digitizer and Memory:

Word size	12 bits
Memory capacity	4096 words
A/D conversion rate	5-MHz max
Memory write rate	5-MHz max
Memory read rate	500 kHz
Aperture uncertainty	± 25 ps

Controller:

Microcomputer type	PDP-11/23
Word size	16 bits
Memory capacity, RAM	16K words
Memory capacity, ROM	20K words
Parallel I/O rate, max	40K wds/s
DMA I/O rate, max	500K wds/s

SYSTEM DESCRIPTION

Referring to the block diagram of Fig. 2, the PMS-01 consists of a TRANSMIT section, RECEIVE section, and CONTROL section.

The TRANSMIT portion of the system consists of a two-channel programmable frequency synthesizer, a power amplifier, and an optional integrator circuit. The synthesizer provides a pulsed sinusoid output on Channel A which is amplified and applied to an acoustical projector. If desired, this toneburst may be integrated before amplification to help reduce spurious responses in the projector. The synthesizer also provides a pulsed TTL output on Channel B which serves as a trigger burst to the digitizer.

The RECEIVE portion of the system consists of a programmable preamplifier and a video ADC with buffer memory. The RECEIVE section is essentially a transient recorder capable of acquiring and storing samples at rates up to 5-million samples per second. These are stored as 12-bit words in a 4K-word buffer memory.

The CONTROL section is a PDP-11/23 microcomputer which serves as a GPIB interpreter, setting up the TRANSMIT and RECEIVE sections in response to commands from the external computer. These commands include such parameters as synthesizer output frequencies and pulse widths, initial buffer memory address, number of words to be digitized, and so on. The CONTROL section also accepts digitized data from the buffer memory, processes it, and sends it to the host computer. The PDP-11/23 program resides in an on-board programmable read-only memory (PROM) which is non-volatile. The system is, therefore, ready for use upon power turn-on and does not require a bootstrap loader.

As was mentioned earlier, the PMS-01 is operated by an external computer or desk-top calculator via the GPIB. Software in this host computer controls the actual progression of measurements performed by the system. A typical PMS-01 measurement sequence is as follows:

(1) A toneburst is produced by the synthesizer, amplified, and sent to the projector, producing sound in the water medium.

(2) The resulting output of a receiver hydrophone is applied to the RECEIVE section preamplifier.

(3) The amplified received signal is applied to the ADC input. The trigger burst produced by the synthesizer is positioned in time to the desired portion of the received toneburst. For each trigger pulse, a sample of the received signal is captured, converted to a 12-bit binary number, and stored in the buffer memory.

(4) When this process is complete, the contents of the buffer memory are transferred to the PDP-11/23 memory for processing and/or transmittal to the host computer.

The above sequence is repeated at each of the signal frequencies of interest. The collected data may be processed to yield the voltage and phase response of the hydrophone versus frequency. Transmitting current and voltage responses (TCR, TVR) are obtained by utilizing other channels of the RECEIVE section preamplifier for application of signals proportional to projector current and voltage. These are digitized and processed along with the hydrophone output signal.

The received signals must be band-limited to avoid aliasing in the digitizing process. This filtering is currently external to the PMS-01. All subsequent filtering of the data can be handled digitally in the host computer. The system, therefore, has a minimum of analog circuitry and is practically adjustment-free.

The three primary system components of the PMS-01 will now be discussed in greater detail. These are: (1) the synthesizer, (2) the receive preamplifier, and (3) the buffered ADC.

Model 35 P/O Coherent Frequency Synthesizer

This device was customized for USRD by Real Time Systems, Inc.* to serve as the signal and ADC trigger source in the PMS-01. It is completely programmable via the GPIB and can also be controlled manually from the front panel. It is pictured in Fig. 3.

As mentioned earlier, the Channel A output is a sinusoid of 1 Hz to 3 MHz in frequency, while the Channel B output is a 1-Hz to 10-MHz TTL pulsetrain. These outputs are independent but are derived from a common master clock, so there is no relative jitter between them. This feature allows the PMS-01 to achieve high accuracy and repeatability in sampling. This is extremely important when processing the sampled data with a discrete Fourier transform (DFT) algorithm. (Phase measurements are particularly sensitive to sample jitter.)

The Channels A and B outputs may be continuous or pulsed, the latter being most common for acoustic measurements. In the pulsed condition, coherent bursts are produced with independently programmable frequencies and burst widths. Furthermore, the Channel B output may be delayed with respect to the Channel A output. This allows the user to account for the sound propagation time through the water before triggering the ADC. The pulse repetition frequency (PRF) may be programmed to a desired value; alternately, an external event can trigger each output sequence. Two additional TTL output signals, called T_O and T_D, are provided. These are gate outputs coincident with the Channels A and B outputs, respectively. Figure 4 shows the synthesizer output signals and associated parameters. The sinusoidal output can deliver 1-V RMS into 50 Ω, and may be attenuated in 0.1-dB steps under computer control. However, if the signal is left

- - - - -

* Real Time Systems, Inc.
152 S. MacQuesten Parkway
Mt. Vernon, New York 10550
(914) 667-0425

unattenuated, the signal-to-noise ratio is optimum. The internal clock in the synthesizer may be locked to an external 1-, 5-, or 10-MHz reference signal, if desired. This synthesizer is now a standard product of Real Time Systems, Inc.

Model 3510A Programmable Wideband Amplifier

This device, also a product of Real Time Systems, Inc., serves as multiplexer and signal conditioner for input signals to the PMS-01 system. Like the synthesizer, this instrument is programmable via the GPIB and may also be set from the front panel. The amplifier is single-ended and has a bandwidth of 1 Hz to 10 MHz. Gain may be programmed from 0 to 60 dB in 1-dB steps, and the output is rated at 1 V RMS maximum into 50 Ω. An 8-channel input multiplexer allows various signal sources to be selected under program control. Channels 0 through 4 are unattenuated while Channels 5 through 7 have internal attenuators of 26, 20, and 12 dB, respectively. This allows higher-than-normal input levels to be applied directly to the amplifier. Input impedance is at least 10 MΩ on the unattenuated channels, and 1 MΩ on the others. A photograph of this instrument is shown in Fig. 5.

Buffered ADC

The buffered ADC is a USRD-built instrument containing a commerically available 5-MHz, 12-bit ADC and a 4K × 12-bit memory. Timing and interface circuitry are also provided in the unit. Figure 6 is a photograph of the device, and Fig. 7 is a block diagram of the circuitry. This drawing also shows the interconnections to the synthesizer and controller.

As shown in Fig. 7, the signal to be digitized is applied to the ADC through an overvoltage protection network. This assures that the device will not be damaged by an excessively large input. ENCODE pulses from the synthesizer Channel B output are applied to the ADC, resulting in a 12-bit conversion for each ENCODE pulse. The resulting binary words are applied to the input of a 4096-word-buffer memory. An address counter determines which memory location is written. ENCODE pulses from the synthesizer are also applied to TTL logic which increments the address counter. This same logic also generates the required control pulses for the memory. In this manner, up to 4096 words of data may be digitized and stored at rates up to 5 MHz.

The ADC is a model MOD-1205 manufactured by Analog Devices of Norwood, MA. It is a single printed-circuit board mounted on a second board containing the overvoltage protection network and I/O connectors.

The buffer memory is implemented with 12-static random-access memory integrated circuits (RAM IC's). These devices are each Type 2147H-1, manufactured by Intel Corp. of Santa Clara, CA. Each IC is organized as 4096 × 1 bit and has a maximum access time of 35 ns.

The address counter is implemented with two Model Am 2940 IC's manufactured by Advanced Micro Devices, Inc., of Sunnyvale, CA. Each IC contains an 8-bit address counter and 8-bit word counter plus additional registers and logic which allow the counters to be programmed. The IC's are cascadable to form larger address and word counts.

The remaining logic is implemented with standard TTL devices and provides the necessary timing and interface functions.

Figure 7 also shows the PDP-11/23 controller which is used to initialize the logic circuitry and accept digitized data from the buffer memory. Three PDP-11/23 interfaces are used: (1) a DRV11 parallel I/O interface is used for logic initialization; (2) a DRV11-B DMA interface transfers the buffer memory contents to the PDP-11/23 memory; and (3) a GPIB 11V-1 interface, made by National Instruments, allows the PDP-11/23 to communicate over the GPIB.

There are three operating modes for the buffered ADC: the SET-UP, WRITE and READ modes. In the SET-UP mode, the synthesizer and ADC operate normally, but nothing is written into the buffer memory. In this mode, the PDP-11/23 initializes the logic to the desired state, predetermining the number of words to be written into (or read from) the buffer memory and the starting memory address. In the WRITE mode, words digitized by the ADC are written into the buffer memory. In the READ mode, the content of the buffer memory is transferred into the PDP-11/23 memory. Note that the synthesizer and ADC board are independent of the TTL logic. The synthesizer is controlled by commands on the GPIB, while the ADC board performs a conversion each time it receives an ENCODE pulse from the synthesizer. The remaining circuitry serves only to handle the buffer memory.

An unusual feature of the MOD-1205 ADC is that no "data ready" output signal is provided for strobing the ADC output into the memory. The ADC has an internal pipeline structure such that three ENCODE pulses are required before the first digitized word appears at the ADC output. Subsequent ENCODE pulses will produce additional data, but the output is always delayed by two words which are in the pipeline. The rule is that output word N will appear at the ADC output 2T + D seconds after application of ENCODE pulse N, where T is the ENCODE period and $250 \leq D \leq 300$ ns. The user must effect his own "data ready" based upon the above rule, but this is not straight-forward since delay D is always greater than 200 ns and, for a 5-MHz conversion rate, ENCODE pulses occur every 200 ns. This problem is overcome in the TTL logic, where precisely timed pulses are developed which strobe the memory at the correct time.

When the contents of the buffer memory are transferred to the PDP-11/23 memory in the READ mode, 16-bit words are formed by appending the lower four address bits to each 12-bit data word. This makes the data easier to sort and is also helpful for debugging purposes. An LED display on the front panel (Fig. 6) monitors the data words and indicates the instrument's status. The ERROR indicator will light if an illegal software operation is performed, such as attempting to enter the READ and WRITE modes simultaneously. Status signals are also transmitted to the controller.

Two momentary pushbutton switches on the front panel allow the user to modify the position of the ENCODE pulses to the ADC while observing them on an oscilloscope. The switches are monitored by the controller which senses their closure and incrementally adjusts the synthesizer Channel B DELAY up or down. In this way, the user can establish the portion of the received waveform to be sampled while observing the signals in real time.

SYSTEM BANDWIDTH AND ACCURACY

The PMS-01 system bandwidth is wide enough to encompass the majority of acoustic measurements at USRD. The transmitting section bandwidth is primarily limited by the choice of power amplifier since the synthesizer covers a 1-Hz to 3-MHz range. The power amplifier installed in the system is an Optimation Model PA250AC which covers a 20-Hz to 100-kHz range; other devices are substituted when working at higher or lower frequencies.

The receiving section bandwidth is limited at the upper end by the MOD-1205 ADC aperture error, to be discussed below. At the lower end, a 1-Hz limit is imposed by the 3510A preamplifier.

System accuracy is limited by three inherent sources of error: analog, quantizing, and aperture errors. These will now be discussed.

Analog error is common to any measuring system or device. It is caused by imperfections in analog circuits, such as gain drift or dc offset voltage. It can and should be reduced to zero by a calibration procedure. A common calibration method is to inject a known input signal and observe the resulting output. This is most effective when it is made a standard part of each measurement sequence. If relative, rather than absolute, levels are of primary importance, then a known attenuator can be used instead of a known signal source. In any event, analog error can be eliminated and is therefore not considered in this error analysis. The other two error types, quantizing error and aperture error, are inherent to the ADC.

Quantizing error exists because the analog input to the MOD-1205 must be encoded into a finite-length binary word. Only 2^{12} input levels can be exactly encoded. Quantizing errors of up to $\pm$ ½ least-significant bit (LSB) will, therefore, occur for other input levels. This is the predominate ADC error at low signal frequencies. It may be treated as a broadband noise superimposed upon the input to an error-free ADC having a peak value of ½ LSB and an RMS value of 1 LSB/$(2\sqrt{3})$.

At very high input signal frequencies, aperture error becomes the primary ADC error source. It is caused by an uncertainty (jitter) in the aperture time of the sample-and-hold (S&H) circuit at the MOD-1205 front end. Aperture error should be held to $\pm$ ½ LSB to prevent a loss of ADC resolution. For the MOD-1205, the maximum aperture uncertainity τ_{au} is $\pm$ 25 ps. For a sinusoidal input signal of frequency f and amplitude E, the greatest aperture error occurs at zero-crossings where the maximum

rate-of-change occurs. This rate is $2\pi fE$ V/s. Therefore,

$$\text{Max. Aperture Error} = \left(2\pi f_{max} E_{max}\right) \tau_{au}$$

$$= \pm \tfrac{1}{2} \text{ LSB (desired)}$$

$$= \pm E_{max}/2^{12}$$

and

$$f_{max} = 1/\left(2\pi \times 2^{12} \times 25 \times 10^{-12}\right)$$

$$= 1.55 \text{ MHz}.$$

Since S&H jitter tends to be random, aperture error is reduced and bandwidth is extended by averaging over multiple signal cycles. Thus, the 1.55-MHz upper limit is a worst-case figure for 12-bit resolution.

SYSTEM PERFORMANCE

To test the system's quality and stability, it was used to make repeated measurements of a sinusoidal test signal. The frequency of the test signal was varied from 1 Hz to 2 MHz; at each test frequency, 200 successive frames of data were collected and reduced to yield repeated estimates of the signal's amplitude and phase. These were then summarized statistically.

Below 50 kHz, each frame of data contained 200 sample points and encompassed two cycles of the test signal. Above 50 kHz, each frame still contained 200 points, but covered an observation time of approximately 40 microseconds (μs), adjusted to encompass an integral number of test cycles. Sampling signals in this manner (over an integral number of test cycles) is called "coherent sampling." This requires that

$$\frac{N_T}{f_T} = \frac{N_S}{f_S}$$

where:

f_T = frequency of test signal

f_S = sampling frequency

N_T = number of periods of f_T in the frame of data

N_S = number of periods of f_S in the frame of data.

These measurement conditions are summarized in Table 1. The test signal source was the system synthesizer.

Each frame of data was a sequence of sample points Y_i, $i = 1, 2, 3, \ldots, N$ where $N = 200$. The TOTAL RMS level was computed as the standard deviation of this sequence,

$$Y_{RMS} = \sqrt{\frac{\sum_{i=1}^{N} (Y_i - \bar{Y})^2}{N}}$$

where

$$\bar{Y} = \frac{1}{N} \sum_{i=1}^{N} Y_i .$$

The amplitude and phase of the test signal $y(t) = \sqrt{2}\, A \cos(\omega t + \phi)$ were estimated using the Discrete Fourier Transform (DFT):

$$F_K = \frac{1}{N} \sum_{i=0}^{N-1} Y_i \exp\,(-j2\pi iK/N)$$

where F_K is a complex number representing the Kth spectral line of the DFT. The RMS amplitude and phase of $y(t)$ are then

$$A = \sqrt{2\left[\mathrm{Re}^2(F_K) + \mathrm{Im}^2(F_K)\right]}$$

and

$$\phi = \tan^{-1}\left[\frac{\mathrm{Im}(F_K)}{\mathrm{Re}(F_K)}\right]$$

with $K = N_T$.

Table 1 illustrates that the measured signal amplitude decreased with increasing frequency (column 4). This was, in fact, due to a rolloff in the signal source, demonstrated by the analog measurements of column 5. Harmonic distortion was also derived from the data and is given in column 6.

The value of coherent sampling with minimal jitter is demonstrated clearly by the repeatability of the phase estimates obtained in this test and summarized in Table 2. The RMS scatter in the phase estimates was 1.3 millidegrees for a 1-Hz test signal and 20.5 millidegrees for a 2-MHz test signal. Similarly, excellent repeatability was obtained in the amplitude estimates, as summarized in Table 3. The RMS scatter in the amplitude measurements was 0.004% at 1 Hz and 0.014% at 2 MHz.

Table 1 -
Measurement Conditions

TEST FREQ. f_T, kHz	SAMPLE FREQ. f_S, kHz	N_T, NO. OF PDS OF f_T PER DATA FRAME	ADC OUTPUT, dB RE FULL-SCALE	TEST SIGNAL AMPLITUDE*, dB, NORMALIZED	HARMONIC DISTORTION, PERCENT
0.001	0.100	2	-0.50		0.09
0.003	0.300	2	-0.50		0.09
0.010	1.000	2	-0.50		0.09
0.033	3.300	2	-0.50		0.09
0.100	10.000	2	-0.50		0.09
0.333	33.300	2	-0.51		0.13
1.000	100.000	2	-0.52		0.12
3.333	333.300	2	-0.53		0.12
10.000	1000.000	2	-0.53		0.18
33.333	3333.300	2	-0.52		0.19
100.000	5000.000	4	-0.55	0.00	0.17
333.333	4761.900	14	-0.63	-0.08	0.39
1000.000	5000.000	40	-1.11	-0.56	1.00
2000.000	5000.000	80	-1.79	-1.25	1.11

*Analog Measurement

Table 2 -
Phase Stability

TEST FREQUENCY f_T, kHz	RMS SCATTER, MILLIDEGREES	MAXIMUM POSITIVE DEVIATION FROM MEAN, MILLIDEGREES	MAXIMUM NEGATIVE DEVIATION FROM MEAN, MILLIDEGREES	SAMPLING FREQUENCY f_S, kHz
0.001	1.3	3.2	4.3	0.100
0.003	1.3	3.6	3.9	0.300
0.010	1.3	3.4	2.7	1.000
0.033	1.4	4.4	3.6	3.300
0.100	4.4	7.5	10.7	10.000
0.333	1.5	4.6	3.6	33.300
1.000	1.3	3.8	4.1	100.000
3.333	1.3	4.1	3.5	333.300
10.000	1.5	3.5	4.2	1000.000
33.333	1.4	3.3	3.7	3333.300
100.000	2.1	5.2	6.1	5000.000
333.333	4.1	14.7	10.7	4761.900
1000.000	1.2	40.7	25.9	5000.000
2000.000	20.5	51.5	44.5	5000.000

Table 3 -
RMS Amplitude Stability

TEST FREQUENCY f_T, kHz	RMS SCATTER, PERCENT	MAXIMUM POSITIVE DEVIATION FROM MEAN, PERCENT	MAXIMUM NEGATIVE DEVIATION FROM MEAN, PERCENT	SAMPLING RATE f_S, kHz
0.001	0.004	0.008	-0.011	0.100
0.003	0.003	0.008	-0.010	0.300
0.010	0.006	0.037	-0.014	1.000
0.033	0.005	0.036	-0.012	3.300
0.100	0.008	0.019	-0.019	10.000
0.333	0.006	0.012	-0.017	33.300
1.000	0.010	0.021	-0.019	100.000
3.333	0.015	0.056	-0.022	333.300
10.000	0.010	0.022	-0.019	1000.000
33.333	0.010	0.024	-0.018	3333.300
100.000	0.010	0.019	-0.022	99.010
333.333	0.010	0.023	-0.020	330.033
1000.000	0.010	0.022	-0.022	990.099
2000.000	0.014	0.036	-0.036	1980.198

The sampling rate is chosen to effect a 2 cycle measurement window with 100 points per cycle. Above 50 kHz this is done through undersampling, where the sampling interval is longer than the test frequency period.

FUTURE PLANS

The PMS-01 is a prototype for future measuring systems at USRD. Several improvements can be made in future systems while retaining the basic design of Fig. 7. One such improvement would be the addition of two more ADC's and associated memories, thus creating three high-speed channels. These would be dedicated to projector voltage (E), projector current (I), and hydrophone response (R), respectively. A second pulse output would be added to the synthesizer to accommodate the additional ADC's. One pulse output would be used for triggering the current and voltage ADC's. The other pulse output would be delayed and would trigger the hydrophone response ADC. The advantage of having three ADC's is that only one toneburst would be required at a given frequency to capture samples of E, I, and R. Three tonebursts are required in the present PMS-01 system, as E, I, and R signals are multiplexed through the 3510 preamp.

Another improvement would be the addition of hardware to perform a fast RMS computation on the ADC output and display this on a front-panel meter, thus creating a gated voltmeter. In the present system, RMS and DFT algorithms are implemented in software only.

Some means of self-calibration will be implemented for the present system as well as future ones. For example, a known signal level may be substituted for the normal system inputs, allowing analog errors to be calibrated out.

Experience with the system will, no doubt, lead to other future additions and improvements.

SUMMARY

The PMS-01 is a complete acoustical measuring system employing a buffered 12-bit video ADC and programmable signal source. Its bandwidth is wide enough to cover the majority of acoustic work. System control is handled by a PROM-programmed microcomputer which communicates with a remote computer via the GPIB. Up to eight signal sources can be sampled and processed under computer control. This report has provided an overall description of the system. Additional details may be obtained by directly contacting the USRD.

REFERENCES

1. L.G. Beatty, J.D. George, and A.Z. Robinson, "Use of the Complex Exponential Expansion as a Signal Representation for Underwater Acoustic Calibration," J. Acoust. Soc. Am. 63, No. 6, 1782-94, June 1978.

2. D.H. Trivett, "A Modified Prony Method Approach to Echo-Reduction Measurement of Time-Limited Transient Signals," NRL Memo. Report 4172, June 1980.

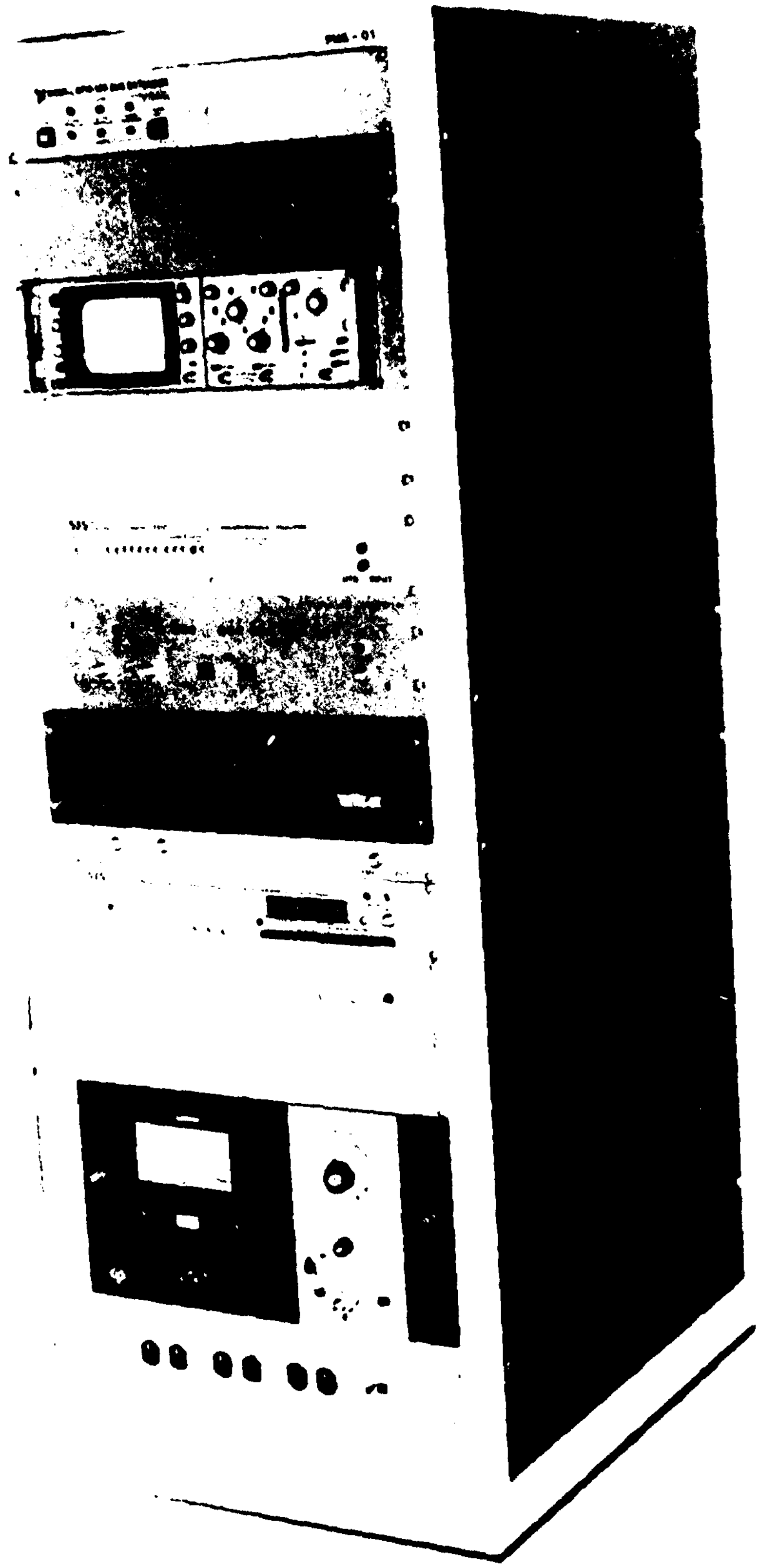


Fig. 1 - PMS-01 System
(USRD Photo 1-3682-11)

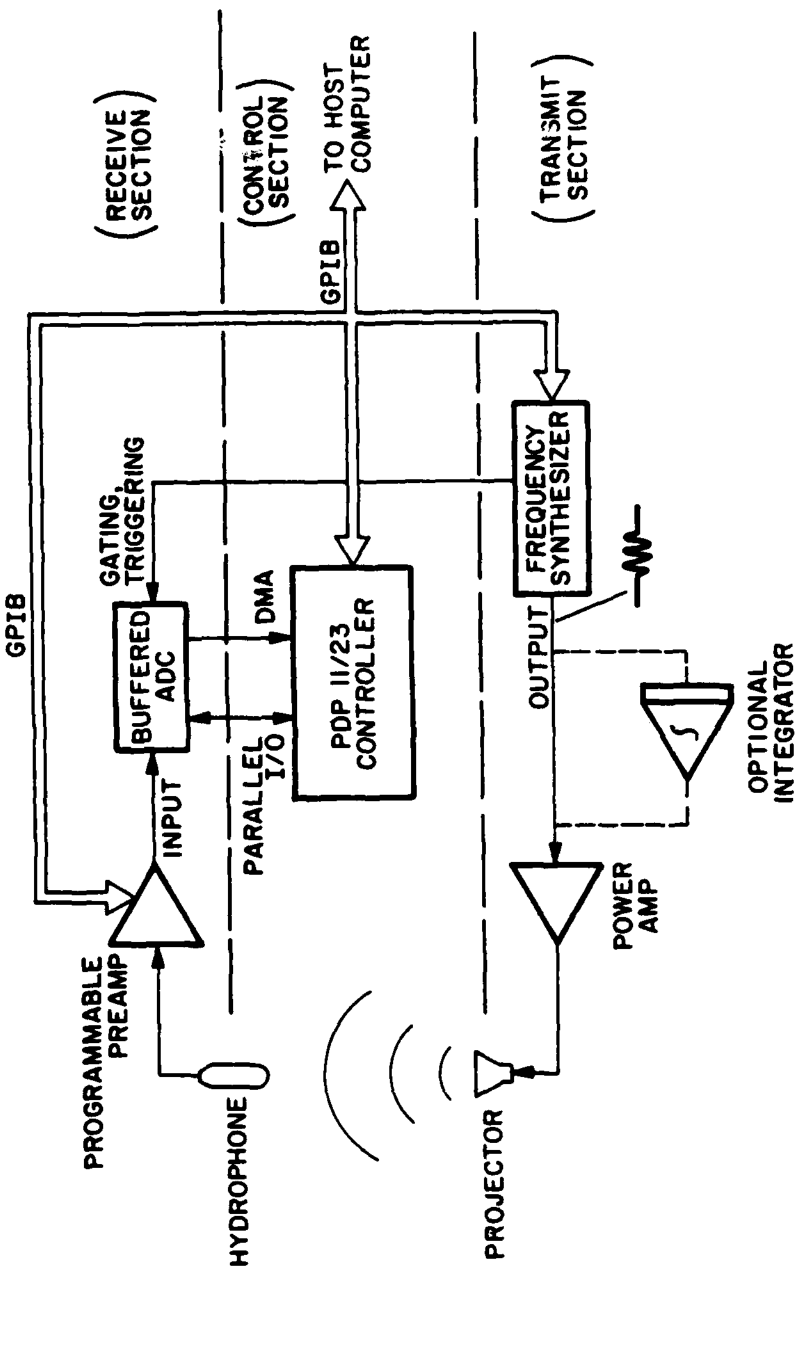


Fig. 2 - PMS-01 Block Diagram

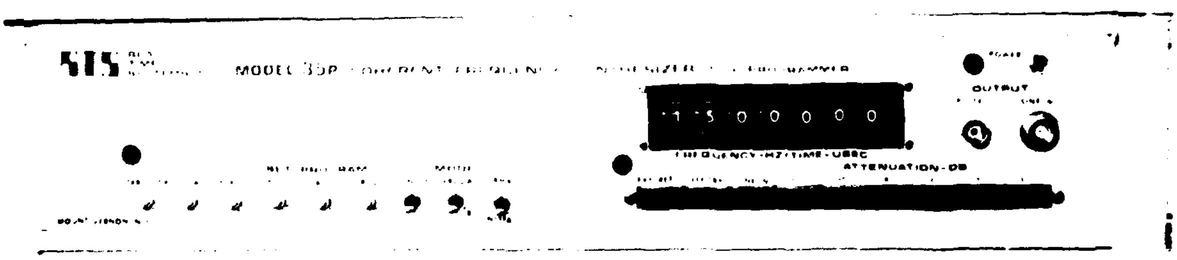


Fig. 3 - Coherent Frequency Synthesizer
(USRD Photo 1-3682-2)

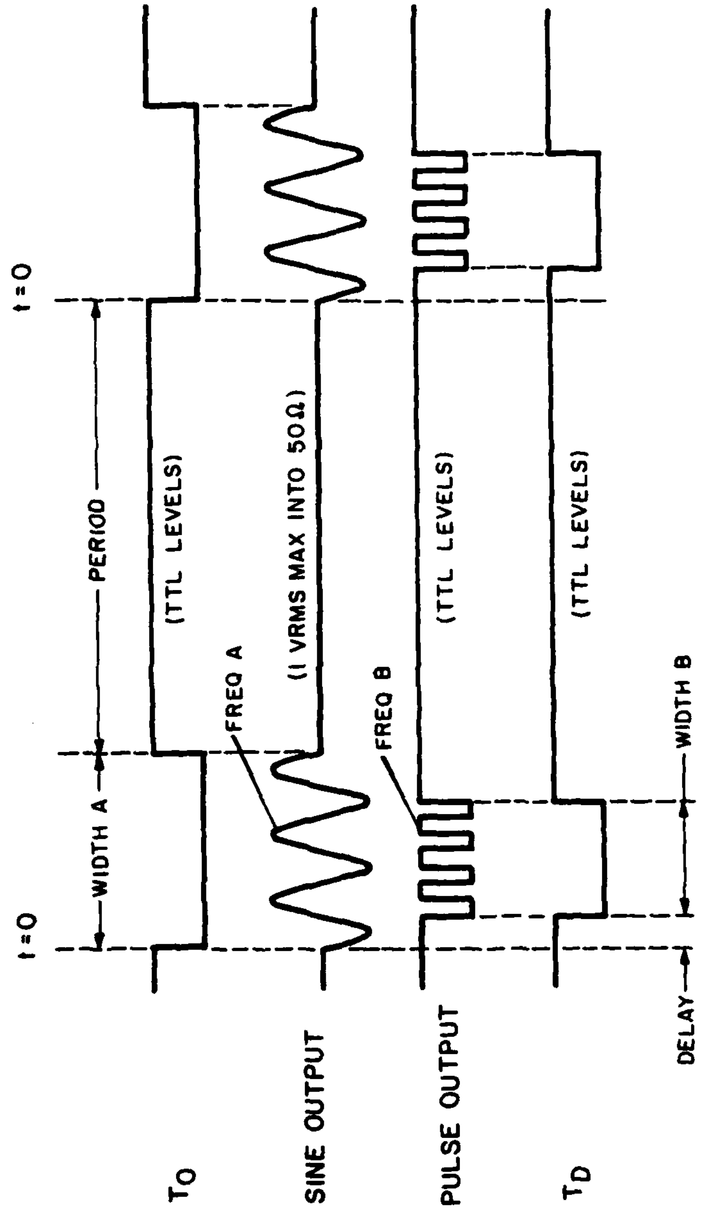


Fig. 4 - Synthesizer Output Waveforms and Parameters

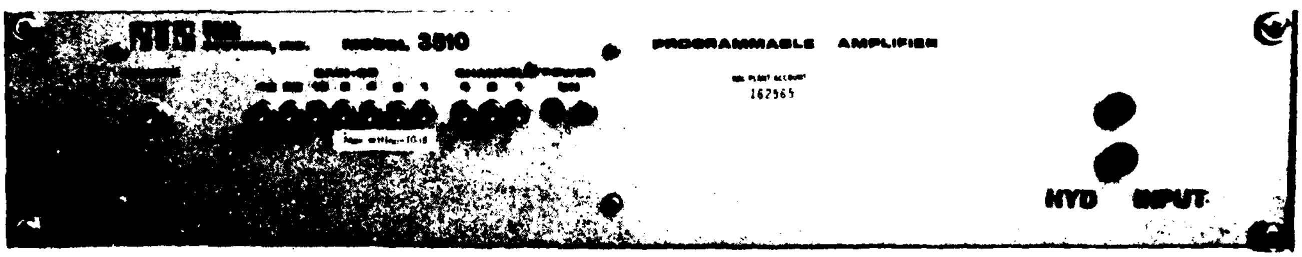


Fig. 5 - Programmable Wideband Amplifier

(USRD Photo 1-3680-17A

Fig. 6 - Buffered ADC
(USRD Photo 1-3680-17A)

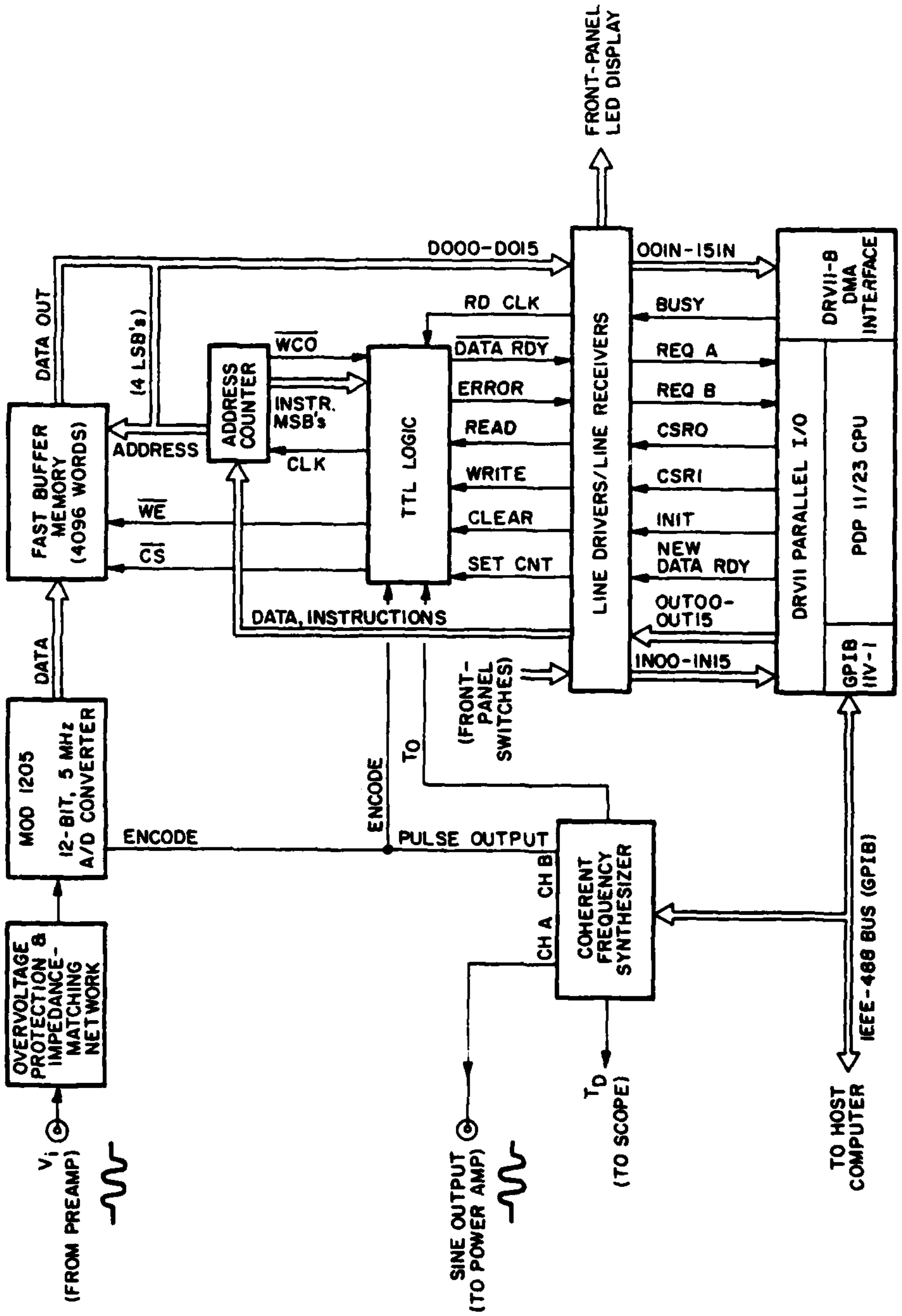


Fig. 7 - Block Diagram of Buffered ADC

DAT
ILM